

IN THE SPECIFICATION:

At page 1, after the title and prior to line 4, please the following new paragraph:

Cross-Reference to Related Applications

This application is the U.S. National Stage of International Application Number PCT/FI2004/050045 filed April 22, 2004 which was published in English on November 4, 2004 under International Publication Number WO 2004/095701 and claims priority under 35 U.S.C. §119 to Finnish Patent Application No. 20035050 filed on April 22, 2003.

At page 1, please amend the paragraphs beginning on line 25 through page 3, line 9 as follows:

In prior art there are many different filter designs for different signal filtering purposes. The filters can be divided into different categories *e.g.* on the basis of the impulse response of the filters. The filters can have either infinite impulse response (IIR) or finite impulse response (FIR). The filters can further be ~~categorised~~categorized into sub categories on the basis of other properties of the filters. In this patent application the finite impulse response filters, or FIR filters, are considered in greater detail.

The finite impulse response of the FIR filters means that if an impulse is input to the FIR filter the output of the FIR filter will stabilize to zero or to a constant value in the course of time. In other words, the effect of the input impulse to the output of the FIR filter is finite in time.

In the following, some terms typical to filters are defined. The filters typically have a certain frequency response. This means that different frequency components of the input signal are attenuated or amplified differently, *i.e.* the

frequency properties of the input signal affect ~~on~~ how the signal passes through the filter. For example, filters ~~having~~having a low-pass frequency response attenuate high frequency signals more than low frequency signals. High-pass filters attenuate low frequency signals more than high frequency signals. Band pass filters have a certain, band pass frequency region on which signals are attenuated less than signals outside the band pass frequency region. Band stop filters have a certain, band stop frequency region on which signals are attenuated more than signals outside the band pass frequency region. The frequency on which the filtering properties change (e.g. from stop band to pass band or *vice versa*) is called as a cut off frequency. Typically the cut off frequency is defined as a frequency on which the attenuation of the filter is 3 dB above the minimum attenuation (or amplification is 3 dB below maximum amplification) of the pass band of the filter. In band pass filters there are two cut off frequencies defined, wherein the pass band lies between the lower cut off frequency and the upper cut off frequency. It should be noted here that in practical implementations the filtering properties ~~does not~~do not change suddenly at the cut off frequency but there is always a transition region in which the attenuation (or amplification) properties of the filter changes. ~~It is~~It will also ~~obvious~~be evident that the frequency response is not necessarily constant on the pass band or on the stop band but there can exist some variations (ripple) as is known by ~~an expert~~anyone of skill in the field~~the art~~.

There are many ways to implement apparatuses containing FIR filters. In some designs adaptivity has been achieved by using some adaptive blocks in the filtering apparatus. As an example of such a filtering apparatus an adaptive interpolated FIR filter, or AIFIR filter for short, is presented in the following. AIFIR filters, which contain one or more interpolators, are applicable in such applications in which a large adaptive FIR filter is required. For example, in echo cancellation, there is a necessity to use a large FIR adaptive filter to model the echo path. When an AIFIR filter is used in a filtering apparatus, this gives an important reduction of the arithmetic operations for both filtering and weight updating. The AIFIR filters are well known by an expert in the field. It should be noted that the interpolator plays an important role in the performance of these structures. The existing approaches in the field of AIFIR filtering apparatuses ~~does not~~do not deal

with the design of the interpolator. There are many applications, such as system identification and channel equalization, in which prior information about the frequency response of the system to be modelled is not available. Therefore, in these applications it is not possible to design a fixed interpolator.

At page 5, please amend the paragraph beginning on line 23 as follows:

According to the Equations (10) and (11), it can be seen that the Equation (7) is equivalent with the update equation of the standard LMS, in which just $N - K$ coefficients are adapted provided that the vector $W(n)$ is ~~initialised~~initialized with zeros. Therefore, the multiplication with F and the addition of q does not introduce extra computations in the Equation (7).

At page 6, please amend the paragraph beginning on line 30 through page 7 line 13 as follows:

The frequency response of the optimum filtering apparatus of the first implementation is presented in Fig. 2, and the frequency response of the AIFIR filtering apparatus according to the first implementation is depicted in Fig. 3 (prior art). Respectively, the frequency response of the optimum filtering apparatus of the second implementation is presented in Fig. 4, and the frequency response of the AIFIR filtering apparatus according to the second implementation is depicted in Fig. 5 (prior art). Now, when the Figs. 2 and 3 are compared, it can be seen that the prior art AIFIR filter of Fig. 3 works quite well, in the case when the frequency response of the interpolator is appropriately chosen (for example, a low-pass interpolator for a low-pass optimum filter). In the case when the design of the frequency response of the interpolator does not match the frequency response of the optimum filtering apparatus (for example, a low-pass interpolator for a high-pass optimum filter) the prior art AIFIR filter totally fails as can be seen when comparing the Figs. 4 and 5.

At page 7, please amend the paragraphs beginning on line 28 through page 8, line 22 as follows:

According to a first aspect of the present invention there is provided a method for filtering comprising adaptive filtering an input signal, interpolating the filtered signal, interpolating the input signal for adapting the adaptive filtering, and adapting the properties of the interpolation of the filtered signal. ~~according to the present invention is mainly characterized by that the properties of the interpolation of the filtered signal are adaptable.~~

According to a first aspect of the present invention there is provided an apparatus comprising an adaptive filter for filtering an input signal; a first interpolator for interpolating the filtered signal; a second interpolator for interpolating the input signal, wherein the interpolated input signal is arranged to be used to adapt the adaptive filter; and a first adapting block for adapting the properties of the first interpolator. ~~according to the present invention is mainly characterized by that the apparatus further comprises a first adapting block for adapting the properties of the first interpolator.~~

Significant advantages are achieved with the present invention. In applications where a very large FIR filter is required, the complexity of the apparatus can be reduced due to the fact that a small number of coefficients are different from zero. Therefore, less calculation operations are needed than with prior art filtering apparatuses. The invention is also applicable with applications in which ~~there~~ it is not possible to have information about the frequency response of an optimum filtering apparatus. Therefore, by using the method of the present invention the frequency characteristics of the apparatus can be adjusted according to the desired frequency response. Also, when there is a need to change the frequency response of the apparatus during operation it is possible with the apparatus of the present invention. The memory space needed to store the filter coefficients is also smaller than with prior art FIR filters.

At page 9, please amend the paragraphs beginning on line 17 through page 10, line 28 as follows

Figs. 9a to 9d depict some of main ~~applications~~application classes as simplified block diagrams.

Detailed Description of the Invention

In Fig. 6 there is presented a block diagram of an apparatus 1 according to an advantageous embodiment of the present invention. The apparatus 1 includes a signal processing block having an adjustable interpolator. The signal processing block is advantageously an adaptive FIR filter 2 in which the input signal $x(n)$ is filtered. Hence, the apparatus 1 according to this advantageous embodiment of the present invention can also be called as an AIFIR filtering apparatus. It is ~~obvious~~should be evident that also other signal processing blocks than FIR filters can be used with the present invention. For example, infinite impulse response filters (IIR) can be used in some applications. The output signal $y(n)$ of the adaptive FIR filter 2 is directed to a first adaptive interpolator 3 and to a first adapting block 4. The interpolated signal is directed from the output of the first adaptive interpolator 3 to the first input 5.1 of a combiner 5. The second input 5.2 of the combiner 5 receives a reference signal $d(n) + z(n)$, which ~~consists of~~includes the desired signal $d(n)$ and noise $z(n)$. The combiner 5 subtracts the output signal from the reference signal of the first adaptive interpolator 3 to form an error signal $e(n)$. The error signal $e(n)$ is directed to the first adapting block 4 and to a second adapting block 6. The first adapting block 4 uses the error signal $e(n)$ and the output signal $y(n)$ of the adaptive FIR filter 2 to form adapting information for the first adaptive interpolator 3. The first adapting block 4 uses the adapting information to change the properties of the first adaptive interpolator 3 when necessary, for example, by changing one or more coefficients of the adaptive interpolator 3. The apparatus of Fig. 6 also comprises a second adaptive interpolator 7 which receives the input signal $x(n)$ and interpolates it to form an interpolated input signal $x_i(n)$. This is necessary in order to have signals with substantially the same sample rate at both inputs of the second adapting block 6. In addition to the error signal $e(n)$, the second adapting block 6 also receives the interpolated input signal

$x_i(n)$. The second adapting block 6 uses the received signals $e(n)$, $x_i(n)$ to change the properties of the adaptive FIR filter 2 when necessary.

In the following, the operation of the individual blocks of the apparatus 1 will be described in more detail. The adaptive FIR filter 2 is sparse FIR adaptive filter having $(L - 1)$ zeros between non-zero coefficients. The coefficients of the adaptive FIR filter 2 are preferably adapted such that the expected value of the squared error is minimized. In order to handle the sparse nature of the adaptive FIR filter 2 a constrained approach has to be used. The constrained cost function to be minimized is the same as with prior art filters. Therefore equations (1) and (2) are applicable here. Then, the similar steps ~~than~~ with as with prior art can be applied as follows:

At page 11, please amend the paragraphs beginning on line 27 through page 12, line 22 as follows:

The ~~behaviour~~behavior of the apparatus according to the present invention can be ~~analysed~~analyzed e.g. by using ~~the similar~~ example situations ~~than~~similar to what was used above in the description where the background art was considered. The frequency response of the optimum filtering apparatus for the first example is depicted in Fig. 2 and the respective frequency response of the apparatus 1 according to the present invention is depicted in Fig. 7. The frequency response of the optimum filtering apparatus for the second example is depicted in Fig. 4 and the respective frequency response of the apparatus 1 according to the present invention is depicted in Fig. 8. It can be seen by comparing the Figs. 2, 3 and 7 that the filtering apparatus 1 according to the present inventions works substantially as well as the prior art filtering apparatus designed properly according to the requirements of the special situation. In that case, both the prior art filtering apparatus and the filtering apparatus of the present invention approximate very well the optimum filter.

In the case when the interpolator of the prior art filtering apparatus is not designed appropriately, the prior art filtering apparatus fails to find optimal coefficients for the adaptive FIR filter. ~~The~~In contrast, the filtering apparatus

1 according to the present invention has also—in this case a very good performance. This can be seen by comparing the Figs. 4, 5 and 8.

Although the apparatus of Fig. 6 comprises the first 3 and the second adaptive interpolators 7, it is ~~obvious~~should be evident that they can be implemented as a single functional unit or a piece of code of a digital signal processor (not shown). If, however, there are two adaptive interpolators 3, 7, they both can (and should) still use the same coefficients. Therefore, there is no need to store the coefficients for the adaptive interpolators 3, 7 twice. This also reduces the memory requirements of the apparatus 1.

At page 12, please amend the paragraph beginning on line 30 through page 13, line 4 as follows:

There are many application areas in which the filter according to the present invention can be applied. Figs. 9a to 9d depict some of the main ~~applications~~application classes as simplified block diagrams. Fig 9a depicts how the apparatus of the present invention comprising double adaptive interpolating FIR filter (DAIFIR) can be used in identification applications. The notion of a mathematical model is fundamental to sciences and engineering. For Applications~~applications~~ dealing with identification the filtering apparatus 1 is used to provide a linear model that represents the best fit to an unknown plant. The plant 8 and the filtering apparatus 1 are provided with the same input signal $x(n)$. The plant output supplies the desired response $d(n)$ for the filtering apparatus 1. If the plant is dynamic in nature, the model will be time varying.

At page 13, please amend the paragraphs beginning on line 26 through page 14 line 27 as follows:

The fourth class of applications is interference modelling and it is depicted in Fig. 9d as a simplified block diagram. In this class of applications, the filtering apparatus 1 is used to cancel unknown interference contained ~~in a~~in a primary signal, with the cancellation being ~~optimised~~optimized in some sense. The primary signal serves as the desired response for the filtering apparatus 1. A reference signal is employed as the input to the adaptive

filtering apparatus. The reference signal is derived from a sensor or set of sensors located in relation to the sensor(s) supplying the primary signal in such a way that the information-bearing signal component is weak or essentially undetectable.

The above described application classes are known by ~~an expert~~anyone of skill in the ~~field~~art of adaptive filters. The present invention provides improved filtering ~~method~~methods to be applied e.g. in those application areas. The improvements are mainly based on the adapting nature of the interpolators, which has not been used with prior art filtering methods.

The above mentioned filtering applications can be utilized, for example, in ~~analysing~~analyzing properties of systems such as buildings, earth, the human body, communication channels, etc. For example, in the case of ~~analysing~~analyzing buildings the input signal can be a shock wave, wherein the filter coefficients can be used in evaluating the ~~behaviour~~behavior of the building during earthquakes.

The filtering method of the present invention can also be used for noise cancellation e.g. to suppress ~~maternal~~maternal ECG component ~~in~~in a fetal ECG. The input signal $x(n)$ of the filtering apparatus 1 is taken near the mother's heart to generate as ~~clean~~clean a heartbeat signal as possible of the mother's heartbeats. The desired signal $d(n)$ is taken near the ~~abdominal~~abdomen of the mother to get a fetal ECG signal. The "error" signal $e(n)$ of the filtering apparatus 1 is then the fetal ECG signal from which the mother's heartbeat signal is substantially totally removed.

It is also possible to use the filtering method of the present invention in channel equalization, time delay estimation, echo cancellation, adaptive ~~control~~control, etc. It is ~~obvious~~should be evident that the above mentioned applications are just non-restrictive examples in which the present invention can be applied.